A Two-Phase Loss Differentiation Algorithm for Improving TFRC Performance in IEEE 802.11 WLANs

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***Abstract*— In IEEE 802.11 WLANs, packet losses may be due to buffer overflow, transmission errors, or collisions. Therefore, the performance of TCP-Friendly Rate Control (TFRC) in IEEE 802.11 WLANs largely depends on its ability to differentiate packet losses resulting from network congestion (due to buffer overflow and collisions) and those from transmission errors. In this paper, an enhanced TFRC (E-TFRC) protocol is proposed to detect and identify the cause of packet loss events through a novel two-phase loss differentiation algorithm (TP-LDA). The packet losses due to buffer overflow and those due to failed transmissions in WLANs are first differentiated. For failed transmissions, the fraction of those due to collisions is obtained with the assistance of the lower layer. By employing TP-LDA, only the packet losses due to buffer overflow and collisions are notified to the sender for appropriate flow and congestion control. To quantify the performance of TFRC and E-TFRC over WLANs, a continuous-time Markov chain based on a new WLAN link model is developed by considering both collisions and transmission errors. Analytical and simulation results demonstrate that, with appropriate loss differentiation, E-TFRC can achieve higher throughput than TFRC in WLANs with different channel profiles.**

***Index Terms*— TCP-friendly rate control, IEEE 802.11 WLANs, two-phase loss differentiation algorithm, WLAN link model, effective throughput.**

I. INTRODUCTION

THE explosive growth of the Internet has been witnessed in the past two decades. The core of the Internet is the simple, robust, scalable IP protocol which provides minimal best effort datagram delivery services. To efficiently and fairly share the network resources in a distributed manner, the end-systems voluntarily deploy the transmission control protocol (TCP) which adjusts the sending rate depending on network conditions. As a result, TCP has been one of the key factors to the success of the Internet. It is expected that the Internet will continuously evolve to incorporate heterogeneous wireless and wired links, and support a wide variety of multimedia applications. For instance, stationary and mobile wireless local area networks (WLANs) based on the IEEE 802.11 standards [1] can provide anytime, anywhere Internet access for both data and multimedia services. However, TCP meets great challenges for supporting multimedia applications in wireless networks. This is because, without explicit knowledge of the causes of packet loss, TCP assumes all packet losses are due to network congestion and it halves the sending rate for any single packet loss, which would be erroneous in wireless networks where packets can be lost due to network congestion or transmission errors. It is therefore imperative to identify the events that cause packet losses in order to properly enforce flow/congestion control with quality of service (QoS) provisioning.

On the other hand, to better support multimedia applications in IP networks and maintain network stability and integrity, TCP-friendly flow/congestion control has been an active research topic [2]. TCP-friendliness can be achieved when the long term throughput of a non-TCP controlled flow does not exceed that of any TCP flows under the same circumstance [3]. A representative TCP-friendly protocol is the TCP-Friendly Rate Control (TFRC) protocol [4]. A TFRC sender sets the sending rate according to a TCP response function [5], which is a function of round trip time, retransmission timeout, packet size, and packet loss event rate. To determine the TFRC sending rate, the TFRC receiver sends feedback messages to the TFRC sender at least once per round trip time (RTT) or whenever a packet loss event is detected. Since TFRC is less aggressive in probing for available bandwidth and more moderate in responding to transient network congestion, the TFRC flow throughput is much smoother than that of the TCP flow. In addition, for time-sensitive applications, the TFRC sender does not need to retransmit lost packets, because end-to-end retransmissions will introduce excessive delay.

However, similar to TCP, TFRC also mis-interprets packet losses due to transmission errors in the wireless domain as congestion indicators. Hence, TCP and TFRC will unnecessarily decrease the sending rate, which results in the under-utilization of the wireless link. To address this problem, several losses differentiation algorithms have been proposed [6]- [9]. Most of them are targeted for TCP/TFRC in wireless cellular

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systems, where a dedicated wireless channel is allocated to a mobile user. So packet losses in the wireless domain can be assumed due to transmission errors, and should not be treated as congestion indicators. However, since IEEE 802.11 WLANs [1] employ a contention-based medium access control (MAC) protocol, a transmission may fail due to either transmission errors or collisions. Packet losses due to collisions are indeed indicators of congestion in WLANs. Thus, ignoring packet losses due to collisions in WLANs may lead to severer congestion and degraded resource utilization in WLANs. In addition, a link layer retransmission scheme is deployed in WLANs to retransmit failed packets up to certain times. A failed transmission after a number of attempts may be due to both collisions and transmission errors in WLANs. It is difficult to differentiate packet losses due to collisions and those due to transmission errors. To the best of our knowledge, how to appropriately interpret packet losses in WLANs for TCP and TCP-friendly congestion control has not been well addressed in the literature.

In this paper, we propose an enhanced TFRC (*E-TFRC*) protocol that is tailored for IEEE 802.11 WLANs. A novel *two-phase loss differentiation algorithm (TP-LDA)* is employed in E-TFRC: packet losses due to buffer overflow are distinguished from those due to failed transmissions; then, using the information from the MAC layer, the fraction of failed transmissions due to collisions in WLANs is estimated. Packet losses due to buffer overflow and collisions are notified to the sender as network congestion indicators. On the other hand, packet losses due to transmission errors are not considered as network congestion indicators. To evaluate the protocol performance, we develop a WLAN link model considering both collisions and transmission errors. Based on the WLAN link model, a continuous-time Markov chain (CTMC) model is developed to evaluate the performances of TFRC and E-TFRC over IEEE 802.11 WLANs. Analytical and simulation results demonstrate that E-TFRC can achieve higher throughput and better QoS than TFRC.

The main contributions of this paper are three-fold: 1) we introduce TP-LDA to improve the TFRC performance by using the measured information at the MAC layer for calculating the packet loss probability in TFRC; 2) we develop an analytical model for the IEEE 802.11 distributed control function (DCF), which captures both collisions and transmission errors due to poor channel condition (e.g., fading, shadowing). A CTMC model is further developed to study the performance of TFRC over IEEE 802.11 WLANs; 3) through numerical analysis and simulations, we investigate the effects of the number of contending nodes in a WLAN, the wireless channel profile, and the access point (AP) buffer size on the TFRC performance. These results demonstrate the relationship between the lower layer protocol parameters and the upper layer protocol performance. Therefore, they can be readily used to further enhance multimedia services in IEEE 802.11 WLANs through a cross-layer approach.

The remainder of the paper is organized as follows. The system model is described in Section II. Section III proposes E-TFRC and TP-LDA over IEEE 802.11 WLANs. In Section IV, the IEEE 802.11 WLAN link model and the CTMC model for TFRC over IEEE 802.11 WLANs are developed. In Section V, the effective throughputs of TFRC and E-TFRC are evaluated in different network environments. Section VI summarizes the related work, followed by concluding remarks in Section VII.

II. System Model

We consider a TFRC flow between a TFRC sender node (SN) and a receiver node (RN). The WLAN is shared by the RN, contending nodes (CNs), and an AP which is connected to the Internet. The AP has a buffer with a limited size B. All packets destined to or originated from the RN and CNs should transverse the AP, and the AP does not have a higher priority to access the channel. Therefore, the downlink from the AP to other mobile nodes (MNs) is the bottleneck in the WLAN. We focus on the packet losses in the WLAN and thus assume that packet losses in the wired domain is negligible, and the packet delay in the wired domain between the SN and the AP, t_{wired}, is constant. In other words, the end-to-end delay jitter mainly occurs in the WLAN. Also, we assume the transmission of feedback packets from the RN to the SN is error-free.

In a WLAN based on the IEEE 802.11 DCF mode [1], N saturated MNs (i.e., each MN always has a packet to send) share the network. We consider a basic access mode without request-to-send (RTS)/clear-to-send (CTS) exchange, because RTS/CTS is not very useful for infrastructure-based WLANs and it is disabled in most products available in the current market. Since the maximum transmission unit (MTU) in IEEE 802.11 is sufficiently large, no link layer fragmentation is considered. The AP broadcasts beacon frames for time synchronization and identification at fixed intervals, typically every 100 ms. Since the MNs and the AP are synchronized, the MNs can detect whether the scheduled beacon frames arrive on time or not. Various studies [10] have shown that considerable transmission errors are due to channel impairment in WLANs. Therefore, a packet transmission attempt in a WLAN may fail due to either transmission error or collision.

III. E-TFRC: Enhanced TFRC over IEEE 802.11 WLANs

E-TFRC employs a *two-phase loss differentiation algorithm (TP-LDA)* in the end-systems, which combines loss differentiation algorithms at the link and transport layers for cross-layer optimization to improve the performance of transport layer protocols (e.g., TFRC and TCP) over IEEE 802.11 WLANs.

We first clarify the terminologies related to packet loss. With respect to the transport layer, there are two kinds of packet losses: *congestion loss* and *link loss*. A congestion loss represents the case that a packet is discarded at the AP due to buffer overflow. A link loss occurs when a packet is lost during transmission in the WLAN. The link loss may be due to two types of transmission failures: *collision failure* and *transmission error failure*.

Since IEEE 802.11 WLANs support link layer retransmissions, the causes of a link loss may be a combination of channel collisions and transmission errors. For example, if the retry limit is 2, the first attempt may fail due to collision, while the second and third attempts may fail due to transmission errors. Therefore, it is difficult to determine if a packet loss is due to collision or transmission error. Instead, we use the ratio between collision failures and transmission error failures for loss differentiation at the MAC layer, which will be elaborated in Section III-B.

A. Phase I: Congestion Loss vs. Link Loss

The first phase (*phase I*) of TP-LDA is to differentiate link losses from congestion losses. Several schemes have been proposed to address this issue. Among them, a differentiation scheme utilizing a relative one-way trip time (ROTT) is introduced in [11]. ROTT measures the one-way packet delay jitter, and it is shown that loss differentiation based on ROTT works well in a shared WLAN channel [6]. Therefore, we employ the ROTT-based loss differentiation scheme in the proposed TP-LDA.

To illustrate the principle of the ROTT-based loss differentiation scheme, the AP buffer (for downlink transmission) is approximated by an *M/M/1/K* queue [12], where K (the system size) equals B (the AP buffer size) plus one. Since the delay jitter in the wired link is fairly stable, the delay jitter in the WLAN is the dominant factor affecting ROTT. Let λ and μ be the arrival and service rates of the *M/M/1/K* queue, and they are determined by the traffic pattern and the WLAN channel conditions, respectively. In the *M/M/1/K* queue, the average system waiting time, i.e., the sum of queuing delay and service time, is given by

$$L = \frac{1-\rho^{K+1}}{\lambda(1-\rho^K)}\left(\frac{1}{1-\rho} - \frac{1+K\rho^{K+1}}{1-\rho^{K+1}}\right), \quad (1)$$

where $\rho = \lambda/\mu$ is the traffic intensity or the utilization factor of the AP buffer. On the other hand, the probability of buffer overflow can be computed as

$$O = \frac{(1-\rho)\rho^K}{1-\rho^{K+1}}. \quad (2)$$

Figure 1 indicates the relation between the average system waiting time and the buffer overflow probability when the average service time is normalized to unity. It can be seen that the buffer overflow rate increases exponentially with the increase in the average system waiting time. Especially, the buffer overflow rate (or congestion loss rate) increases drastically when the average system waiting time exceeds a certain threshold. For instance, they are 3.2 $(1/\mu)$ and 4.9 $(1/\mu)$ for $K = 20$ and $K = 40$, respectively. Note that the probability of a target packet being lost due to collision or transmission error is independent of how long the packet has been waiting in the AP's queue. In addition, as discussed in [13], the delay in the wired domain is fairly stable. Therefore, the dynamics in ROTT is mainly affected by the buffer overflow probability, and we can infer that a packet loss is mainly due to congestion if there is a spike in ROTT.

In the ROTT-based loss differentiation scheme, two states, NORMAL and SPIKE, are defined. The default state is NORMAL, but the state transition occurs if there is a spike in the estimated ROTT. For state identification, two thresholds, B_{upper} and B_{lower}, are defined as [6]

$$B_{upper} = rott_{\min} + \alpha \times (rott_{\max} - rott_{\min})$$

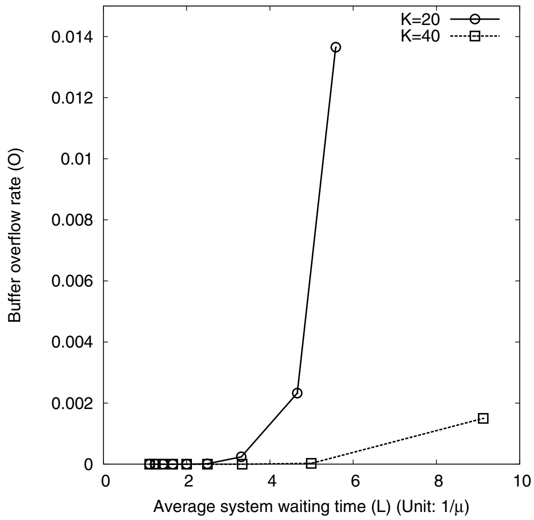


Fig. 1. Buffer overflow rate vs. Average system waiting time.

and

$$B_{lower} = rott_{\min} + \beta \times (rott_{\max} - rott_{\min}),$$

where $rott_{\max}$ and $rott_{\min}$ are the maximum and minimum ROTT measured so far, and $\alpha \geq \beta$. By [6], α and β are set to $1/2$ and $1/3$, respectively. The operations of the ROTT-based loss differentiation scheme are as follows. When a packet is received, the receiver calculates the ROTT of the packet. If the calculated ROTT is larger than B_{upper} and the current state is NORMAL, the state transits to SPIKE. On the other hand, if the ROTT of the received packet in the SPIKE state is less than B_{lower}, the state becomes NORMAL. If a packet loss event is detected and the current state is SPIKE, the packet loss is considered as a congestion loss; otherwise, it is assumed as a link loss.

B. Phase II: Collision Failure vs. Transmission Error Failure

At the second phase (*phase II*), collision and transmission error failures are identified by the receiver using the information from the MAC layer.

Since retransmission is supported in IEEE 802.11 WLANs, a packet loss can be caused by a combination of collisions and transmission errors. In our approach, rather than discriminating two failures strictly, we calculate the ratio between collision failures and transmission error failures by using beacon messages. In infrastructure-based IEEE 802.11 WLANs, beacon messages are broadcasted periodically by the AP. The size of a beacon message is very small and its transmission rate is the basic rate in the IEEE 802.11b specification. From [14], it can be shown that the transmission error probability of a small frame sent at low data rates is negligible. We define a *beacon loss rate (BLR)* as

$$BLR = \frac{T-R}{T}, \quad (3)$$

where R and T are the number of beacon messages received by the MNs, and the number of beacon messages sent by the AP, respectively. The AP schedules a beacon frame in every beacon interval T_{BI}, typically 102.4 ms. According to the IEEE 802.11 specification, a beacon frame is transmitted through channel contention and thus it can be delayed and collided. From [16], the average beacon delay is around 0.5 ms and it is less than 2 ms in most cases. Therefore, the inter-beacon arrival time normally falls in $[T_{BI}, T_{BI}+\theta]$ where θ is the channel contention delay for a beacon message, e.g., θ can be set to 2 ms. The MN monitors the beacon messages: upon receiving a new beacon message, if the inter-beacon arrival time is smaller than $T_{BI}+\theta$, both T and R are increased by one; if the inter-beacon arrival time is larger than $k\cdot(T_{BI}+\theta)$ and smaller than $(k+1)\cdot(T_{BI}+\theta)$ $(k=\{1,2,3,...\})$, T is increased by $(k+1)$ and R is increased by one. Since most beacon losses are caused by channel collisions, BLR can be considered as the collision probability for a link loss. Then, the fraction of collision failures for a link loss, denoted by γ, can be represented by

$$\gamma = BLR, \qquad (0 \le \gamma \le 1). \tag{4}$$

High collision rate indicates that the WLAN is overloaded and congested; therefore, packet losses due to collisions should be counted as congestion indicators. Specifically, γ is first measured at the MAC layer and utilized at the transport layer for calculating the packet loss probability in the TFRC receiver. In the sequel, the updated packet loss probability is fed back to the TFRC sender. Detailed procedures are given below.

C. TP-LDA: Overall Procedure

According to the TFRC protocol specification [15], the TFRC receiver maintains a data structure that keeps track of which packets have arrived and which are missing. A packet loss is detected by the arrival of at least three packets with a higher sequence number than the lost packet. If a packet arrives late (after three subsequent packets arrived) in TFRC, the packet can fill the hole in TFRC's reception record, and the receiver can recalculate the packet loss event rate.

The overall procedure of TP-LDA is as follows. When a packet loss is detected, the TFRC receiver determines whether the packet loss is a link loss or a congestion loss by the ROTT-based scheme. In other words, if the current state is SPIKE, the packet loss is considered as a congestion loss; otherwise, a link loss is assumed. If the packet loss is identified as a congestion loss, the packet loss probability p is updated as the original TFRC, i.e., $C_p \leftarrow C_p + 1$, where C_p is the counter for packet losses. On the other hand, if the packet loss is considered as a link loss, p is updated using γ. Namely, C_p is set to $C_p+\gamma$, where $0 \le \gamma \le 1$. After updating C_p, the packet loss rate p is calculated and notified to the TFRC sender.

IV. Performance Analysis

To evaluate the performance of E-TFRC, we first develop an analytical WLAN link model, which captures correlated transmission errors as well as packet collisions. Based on the link model, we develop the CTMC model for studying the performance of E-TFRC/TFRC over IEEE 802.11 WLANs.

A. IEEE 802.11 WLAN Link Model

The well-known analytical model proposed by Bianchi [17] and its variants [18], [19] do not consider the impact of transmission errors in WLANs. Even though several analytical models [20]- [23] have been proposed to remedy this problem, they focus on random transmission errors without considering the bursty errors due to channel fading, shadowing, etc. However, in WLANs with low user mobility, transmission errors are bursty and highly correlated.

The correlated transmission errors in WLANs can be approximated by using a packet-level discrete-time two-state Markov model[1]. In the Markov chain, time is discretized into virtual slots and a virtual time slot is defined as the time interval between two consecutive backoff counter decrements [2]. The average time slot duration ($E[slot]$) is assumed to be shorter than the channel coherence time and $E[slot]$ can be obtained from Appendix I. Therefore, the channel state remains the same during a virtual time slot and the channel state is either good (g) or bad (b). In the bad state, a packet transmission fails while a packet transmission is successful in the good state if there is no collision. The transition probabilities of these states are given by the matrix

$$\mathbf{Q} = \begin{pmatrix} q_{bb} & q_{bg} \\ q_{gb} & q_{gg} \end{pmatrix}, \tag{5}$$

where q_{xy} is the transition probability from state $x \in \{b,g\}$ to state $y \in \{b,g\}$. Given an average error rate (π_b^{WLAN}), defined as the ratio of the number of erroneous packets to the total number of packets sent (i.e., the probability that the channel in a virtual time slot is in the bad state), and a burst length (l_B), defined as the average number of consecutive virtual time slots in the bad state, the transition probabilities can be computed as

$$q_{bg} = \frac{1}{l_B} = 1 - q_{bb}$$

and

$$q_{gb} = \frac{\pi_b^{WLAN} q_{bg}}{1-\pi_b^{WLAN}} = 1 - q_{gg}.$$

The fading parameters l_B and π_b^{WLAN} can be obtained from measurements or from wireless channel profiles (considering the physical characteristics of the channel and the physical layer modulation and coding schemes used).

Figure 2 illustrates the transmission success/failure behaviors in the IEEE 802.11 basic access mode. A packet transmission fails when the channel is bad (b) or when the channel is good (g) but a collision ($coll$) occurs. A maximum of m retransmissions are attempted before the packet is discarded (i.e., after $m+1$ transmission failures). For every transmission failure, the sender defers its transmission for a random duration, i.e., backoff interval. In Figure 2, τ_{i+1} denotes the transmission epoch of the ith backoff stage $(0 \le i \le m)$, where m is the transmission retry limit. Since the backoff

[1]The two-state Markov model is originated from the Gilbert-Elliot model [24], [25].

[2]In this paper, we simply estimate the duration of an error burst in terms of the average duration of virtual time slot, instead of estimating the number of idle/collision/transmission slots during an error burst. This approximation can still capture the first-order statistics of WLAN channel.

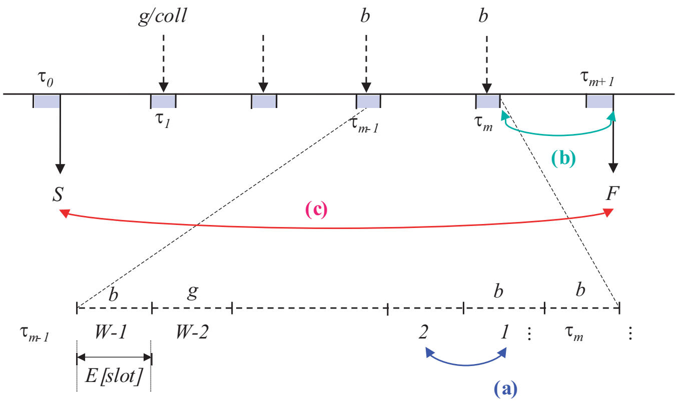


Fig. 2. Transmission success/failure behaviors in the IEEE 802.11 WLANs: (a) a slot, (b) a backoff stage, (c) a packet transmission interval. *coll*, S, and F represent collision, successful transmission, and failed transmission, respectively.

stage length (in numbers of $E[slot]$) is uniformly chosen in the range $[0, W_i - 1]$, where $W_i = 2^i CW_{min}$ and CW_{min} is the minimum contention window size, the average length of the ith backoff stage (n_i) can be computed as $n_i = (W_i - 1)/2$. Then, the transition probability from state $x \in \{b, g\}$ at time τ_i to state $y \in \{b, g\}$ at time τ_{i+1}, denoted by $q_{xy}^{(n_i)}$, can be obtained from elements in the n_i-step transition probability matrix,

$$\mathbf{Q}^{(n_i)} = \begin{pmatrix} q_{bb}^{(n_i)} & q_{bg}^{(n_i)} \\ q_{gb}^{(n_i)} & q_{gg}^{(n_i)} \end{pmatrix}. \tag{6}$$

Since a packet loss occurs after $m + 1$ consecutive transmission failures, there are 2^{m+1} possible unique sequences (traces) of the WLAN channel states for a packet loss event. Let $T_i \in \{(s_0, s_1, ...s_m)\}$ denote the ith trace of all possible traces, where $s_k \in \{b, g\}$ is the WLAN channel state in the kth transmission attempt. Then, the packet loss probability in the WLAN link is given by

$$\pi_F^{WLAN} = \sum_{i=1}^{2^{m+1}} \Theta(T_i) \cdot p_c{}^{N(T_i)}, \tag{7}$$

where p_c is a conditional collision probability when a packet transmission occurs, and $N(T_i)$ is the number of transmission attempts in the good channel state (or the number of packet collisions) in the trace T_i. $\Theta(T_i)$ is the occurrence probability of trace $T_i = (s_0, s_1, ..., s_m)$, and it can be found as

$$\begin{aligned} \Theta(T_i) &= \pi_b^{WLAN} q_{bs_0}^{(0)} q_{s_0 s_1}^{(1)} ... q_{s_{m-1}s_m}^{(m)} \\ &+ \pi_g^{WLAN} q_{gs_0}^{(0)} q_{s_0 s_1}^{(1)} ... q_{s_{m-1}s_m}^{(m)} \end{aligned} \tag{8}$$

where $q_{xy}^{(i)}$ $(x, y \in \{b, g\})$ represents $q_{xy}^{(n_i)}$ in (6). π_b^{WLAN} and π_g^{WLAN} are the steady state probabilities that the WLAN link conditions are bad and good, respectively. π_b^{WLAN} and π_g^{WLAN} are given by $q_{gb}/(q_{gb} + q_{bg})$ and $q_{bg}/(q_{gb} + q_{bg})$, respectively.

To compute the conditional collision probability p_c, we use the same assumptions and notations in [18]. Given N saturated nodes, the conditional collision probability is given by $p_c = 1 - (1 - \tau)^{N-1}$, where τ is the probability that a packet is transmitted in a randomly chosen slot. The packet transmission failure probability p_f in a given slot can be found as

$$p_f = \pi_b^{WLAN} + (1 - \pi_b^{WLAN}) p_c. \tag{9}$$

Let $b(t)$ and $s(t)$ denote the stochastic processes representing the backoff window size and the backoff stage for a given node at time t. The packet transmission behavior can be described by a two-dimensional process as $\{s(t), b(t)\}$. Let $b_{i,k}$ be the stationary distribution of the Markov chain, i.e., $b_{i,k} = \lim_{t\to\infty} P\{s(t) = i, b(t) = k\}$. Then τ is given by

$$\tau = \sum_{i=0}^{m} b_{i,0} = \frac{1 - p_f^{m+1}}{1 - p_f} b_{0,0}, \tag{10}$$

where $b_{0,0}$ is given by (11) and m' is the number of contention window sizes (i.e., the maximum contention window size is $2^{m'}$) and W is the minimum contention window size. By considering (9), (10), and (11), p_c can be computed numerically.

B. CTMC for TFRC over IEEE 802.11 WLANs

Since TFRC is a rate-based protocol, the TFRC sender produces a smooth flow with rate λ (packets per second) that is determined by

$$\lambda = \frac{s}{r\sqrt{\frac{2p}{3}} + 3p(t_{RTO}(1 + 32p^2)\sqrt{\frac{3p}{8}})}, \tag{12}$$

where r is the round-trip time, t_{RTO} is the retransmission timeout value, s is the packet size, and p is the packet loss event rate. The TFRC receiver measures p, while r and t_{RTO} are estimated and calculated by the TFRC sender. Initially, the TFRC sender sets its sending rate to one packet per second and doubles the rate every RTT until a packet loss occurs. Thereafter, the sending rate is determined by (12).

The behavior of TFRC over IEEE 802.11 WLANs can be modeled by a two-dimensional CTMC. In the CTMC, a state is defined as (c, n), where $c \in \{S, F\}$ and $0 \le n \le B$. c represents the transmission result (i.e., successful transmission (S) or failed transmission (F)) of the last packet being sent and n is the number of packets in the AP buffer. The reason that c refers to the state for the last packet is that when a new packet arrives at the buffer, it does not change c, i.e., c is changed only when a packet is processed. The service rate of the AP buffer is determined by the state c. Let μ_{XY} be the service rate from state $X \in \{S, F\}$ to state $Y \in \{S, F\}$. The transition diagram is shown in Figure 3 and the balance equations are summarized in Table I. Since μ_{XY} is the inverse of the average service time denoted by t_{XY}, we need to calculate t_{XY} for μ_{XY}.

First, t_{FF} and t_{SF} are the average service times when a packet is discarded after m retransmissions. Since the backoff interval is independently determined at each backoff stage, t_{FF} and t_{SF} have the same value and they are given by

$$t_{FF} = t_{SF} = \sum_{i=0}^{m} \frac{W_i - 1}{2} \cdot E[slot], \tag{13}$$

where $(W_i - 1)/2$ is the average number of virtual time slots at the ith backoff stage.

$$b_{0,0} = \begin{cases} \frac{2(1-2p_f)(1-p_f)}{W(1-(2p_f)^{m+1})(1-p_f)+(1-2p_f)(1-p_f^{m+1})} & m \le m' \\ \frac{2(1-2p_f)(1-p_f)}{W(1-(2p_f)^{m'+1})(1-p_f)+(1-2p_f)(1-p_f^{m+1})+W2^{m'}p_f^{m'+1}(1-2p_f)(1-p_f^{m-m'})} & m > m' \end{cases}, \quad (11)$$

TABLE I
BALANCE EQUATIONS.

Case	Balance equation
$b=0$	$\lambda \cdot \pi(S,0) = \mu_{SS} \cdot \pi(S,1) + \mu_{FS} \cdot \pi(F,1)$ $\lambda \cdot \pi(F,0) = \mu_{SF} \cdot \pi(S,1) + \mu_{FF} \cdot \pi(F,1)$
$1 \le i \le B-1$	$(\mu_{SS} + \mu_{SF} + \lambda) \cdot \pi(S,i) = \lambda \cdot \pi(S,i-1) + \mu_{SS} \cdot \pi(S,i+1) + \mu_{FS} \cdot \pi(F,i+1)$ $(\mu_{FF} + \mu_{FS} + \lambda) \cdot \pi(F,i) = \lambda \cdot \pi(F,i-1) + \mu_{SF} \cdot \pi(S,i+1) + \mu_{FF} \cdot \pi(F,i+1)$
$b=B$	$(\mu_{SS} + \mu_{SF}) \cdot \pi(S,B) = \lambda \cdot \pi(S,B-1)$ $(\mu_{FS} + \mu_{FF}) \cdot \pi(F,B) = \lambda \cdot \pi(F,B-1)$

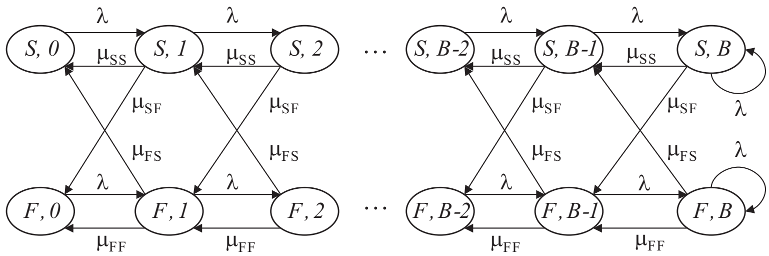


Fig. 3. Transition diagram in CTMC for TFRC over IEEE 802.11 WLANs.

On the other hand, t_{SS} and t_{FS} depend on the backoff stage at which the packet is successfully transmitted. Let Q_j be the jth trace when a successfully transmitted packet reaches the ith backoff stage, where $1 \le j \le 2^i$ and $Q_j \in \{(s_0, s_1, ..., s_{i-1})\}$. The space size of Q_j is 2^i. Then, the probability that a successfully transmitted packet reaches the ith stage when the channel state at the last transmission attempt of the previous packet (i.e., τ_0 in Figure 2) is g, η_g^i, can be computed as

$$\eta_g^i = \sum_{j=1}^{2^i} \Theta_g(Q_j) {p_c}^{N(Q_j)} - \pi_F|g, \quad (14)$$

where $N(\cdot)$ is as defined in Section IV-A. $\Theta_g(Q_j)$ is the occurrence probability of Q_j given that the WLAN link state at the last transmission attempt of the previous packet is a good state. For a given $Q_j = (s_0, s_1, ..., s_{i-1})$, $\Theta_x(Q_j) \triangleq q_{xs_0}^{(0)} q_{s_0 s_1}^{(1)} ... q_{s_{i-2}s_{i-1}}^{(i-1)}$, where $x \in \{b, g\}$. On the other hand, $\pi_F|x, x \in \{b, g\}$ is the packet loss probability when the channel state at the last transmission attempt of the previous packet is x. Similarly, the probability that a successfully transmitted packet reaches the ith stage when the channel state at the last transmission attempt of the previous packet is b, η_b^i, can be obtained from

$$\eta_b^i = \sum_{j=1}^{2^i} \Theta_b(Q_j) {p_c}^{N(Q_j)} - \pi_F|b. \quad (15)$$

For t_{SS}, since the previous packet is successfully transmitted, the channel condition at the last transmission attempt of the previous packet is good. Therefore, t_{SS} is given by

$$t_{SS} = \frac{W_0 - 1}{2} \cdot E[slot] + \sum_{i=1}^{m} \frac{\eta_g^i - \pi_F|g}{1 - \pi_F|g} \cdot \frac{W_i - 1}{2} \cdot E[slot], \quad (16)$$

where $1 - \pi_F|g$ is the probability that a packet is successfully transmitted when the channel condition at the last transmission attempt of the previous packet is good. Note that the first backoff with length $(W_0 - 1)/2$ is always triggered regardless of the transmission result.

On the other hand, for t_{FS}, the last transmission of the previous packet fails due to transmission error (i.e., state b in τ_0) or channel collision (i.e., state g in τ_0). Therefore, both cases should be considered. Let A and B be the probabilities that the last transmission failure is due to channel collision and transmission error, respectively. Then, they are given by $A = \sum_i \Theta(T_i | s_m = g) \cdot {p_c}^{N(T_i|s_m=g)}$ and $B = \sum_i \Theta(T_i | s_m = b) \cdot {p_c}^{N(T_i|s_m=b)}$, where s_m is the WLAN link state at the mth transmission attempt. Then, t_{FS} can be computed as

$$\begin{aligned} t_{FS} &= \frac{W_0 - 1}{2} \cdot E[slot] \\ &+ \sum_{i=0}^{m} \frac{W_i - 1}{2} \cdot \frac{\eta_g^i - \pi_F|g}{1 - \pi_F|g} \cdot \frac{A}{A+B} \cdot E[slot] \\ &+ \sum_{i=0}^{m} \frac{W_i - 1}{2} \cdot \frac{\eta_b^i - \pi_F|b}{1 - \pi_F|b} \cdot \frac{B}{A+B} \cdot E[slot]. \end{aligned} \quad (17)$$

From the CTMC, the packet loss probability p in the TFRC flow can be computed as

$$p = \sum_{c \in \{S,F\}} \pi(c, B) + \sum_{n=1}^{B-1} \pi(F, n), \quad (18)$$

where the first and second terms on the right-hand side represent the packet loss probabilities due to AP buffer overflow and WLAN link loss, respectively. On the other hand, in TFRC with phase I of TP-LDA, only congestion losses are considered for calculating p. Hence, p is given by

$$p = \sum_{c \in \{S,F\}} \pi(c, B). \tag{19}$$

In E-TFRC, p is determined by taking into account congestion losses and γ. Therefore, p can be computed as

$$p = \sum_{c \in \{S,F\}} \pi(c, B) + \gamma \cdot \sum_{n=1}^{B-1} \pi(F, n). \tag{20}$$

The round trip time r of a TFRC flow can be calculated as

$$r = 2t_{wired} + \frac{1}{\lambda} \sum_{n=1}^{B} n \cdot (\pi(S, n) + \pi(F, n)) + t_{up}, \tag{21}$$

where t_{wired} is the one-way delay in the wired link and t_{up} is the latency for WLAN uplink transmission, i.e., from the RN to the AP. Since no packet loss in uplink transmission is assumed, t_{up} can be obtained from

$$\begin{aligned} t_{up} &= \frac{W_0 - 1}{2} \cdot E[slot] \\ &+ \sum_{i=1}^{m} \frac{W_i - 1}{2} \cdot \sum_{i=1}^{2^i} \Theta(Q_j) {p_c}^{N(Q_j)} \cdot E[slot]. \end{aligned} \tag{22}$$

Consequently, the steady state probability $\pi(c, n)$ can be computed from the balance equations in Table I and the normalized condition, $\sum_{c \in \{S,F\}} \sum_{n=0}^{B} \pi(c, n) = 1$. To calculate $\pi(c, n)$, an iterative algorithm is used. First, $\pi(c, n)$ is initialized, with λ_{Init} initialized to 1. r and p are calculated using λ_{Init}, (18) (or (19), (20)) and (21), while t_{RTO} is set to $4r$ [4]. Subsequently, λ and $\pi(c, n)$ are repeatedly calculated until λ converges.

V. Numerical Results

For performance evaluation, we consider three protocols: TFRC, TFRC with the first phase differentiation algorithm only (referred to as TFRC+Phase I), and TFRC with TP-LDA (i.e., E-TFRC). The effective throughput T is defined as

$$T = \frac{\lambda(1 - p)}{r}, \tag{23}$$

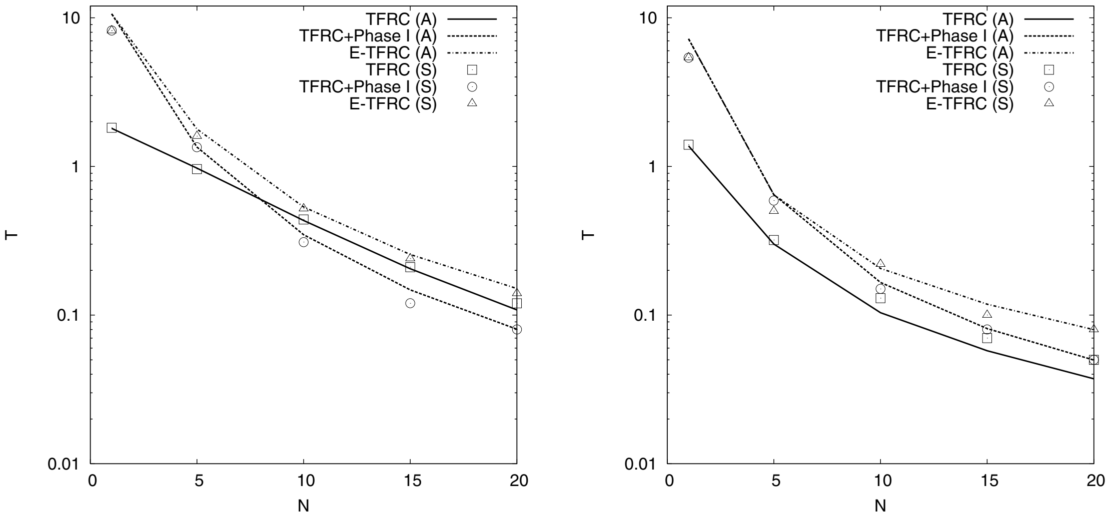
where λ, p, and r are obtained from the CTMC, and it is used as a performance metric. The default IEEE 802.11b values in [18] are used as the WLAN parameters. A TFRC flow is established between the sender node and the receiver node whereas other contending nodes in WLAN are saturated. The packet size in the TFRC flow is 250 bytes, which is smaller than the typical Internet packet size because TFRC is used for multimedia applications with small packet sizes. The default buffer size B is 40 packets and t_{wired} is fixed at 20 ms. To investigate the effects of different burst channel errors and frame error rates (FERs), we define four different WLAN channel profiles: (1) low FER and short burstiness, (2) low FER and long burstiness, (3) high FER and short burstiness, and (4) high FER and long burstiness. π_b^{WLAN} in the low and high FER profiles are set to 0.01 and 0.10, respectively. On the other hand, the burst lengths l_B (in numbers of $E[slot]$) in the short and long burst profiles are 100 and 1000, respectively. To validate the analytical results, extensive simulations are performed using the *ns-2* simulator [26]. In the simulations, we use a continuous-time two-state error model, where the average state sojourn times in the good and bad states are set according to the average state sojourn times in the discrete-time error model used in the analysis.

A. Effect of N

Figure 4 shows the effective throughput T under the low FER. It is observed that as N increases, T is drastically reduced. This is because a large N induces more frequent channel collisions and hence more packets are lost during their transmissions. It can be seen that, in terms of throughput, E-TFRC outperforms TFRC and TFRC+Phase I. This observation can be explained as follows. In TFRC, the sender treats all packet losses as congestion indicators, so that it over-estimates the packet loss rate and sets the sending rate lower than the desired rate. On the contrary, TFRC+Phase I ignores the collision probability when the packet loss rate is computed. Therefore, it under-estimates the packet loss event rate and thus the sender will overshoot the available bandwidth by setting a higher sending rate. Since a higher sending rate makes the WLAN more congested, packet losses due to buffer overflow and collisions become even more severe. Hence, the throughput achieved by TFRC+Phase I is even lower than that of E-TFRC. From these observations, it can be concluded that accurate estimation of packet loss rate due to congestion is very important to fully utilize the available bandwidth in WLANs.

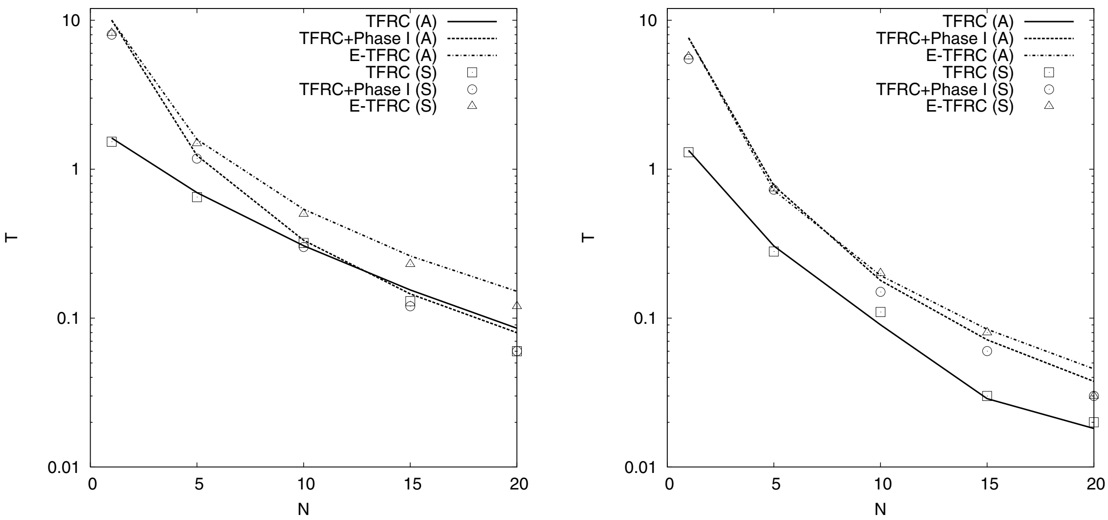
Also, it can be seen that the difference between E-TFRC and TFRC is more significant with channel profile 2 (Figure 4(b)) than that with channel profile 1 (Figure 4(a)). In other words, when the transmission error burst length is long, the effectiveness of TP-LDA is more prevalent. In addition, TFRC+Phase I has an effective throughput similar to that of E-TFRC when N is small. As N increases, the effective throughput of TFRC+Phase I decreases more quickly than that of E-TFRC. When N is small, channel collision is not a dominant factor to affect the effective throughput and thus the impact of the distinction between transmission error and channel collision (i.e., phase II in TP-LDA) is not significant. When N is large, collisions occur more frequently, and under-estimating collisions and overshooting available bandwidth will lead to severer collisions and reduced throughput. Thus, the throughput gain of E-TFRC over TFRC+Phase I is more remarkable for a large N. Consequently, it can be concluded that TP-LDA is more preferable in error-prone and heavily loaded WLANs.

Figure 5 shows the effective throughput in a WLAN with high FER. It is shown that the overall trends are similar to those in Figure 4. However, the throughput gain of E-TFRC to TFRC is more significant when FER is high. This is because TFRC cannot differentiate congestion loss and link loss. When channel condition is worse, the throughput degradation in TFRC becomes more apparent due to the unnecessary sending rate decrease. On the other hand, if FER is high and burstiness is long (i.e., channel profile 4), transmission error is a main contributor to a link loss rather than channel collision.



(a) 1: Low FER and short burstiness (b) 2: Low FER and long burstiness

Fig. 4. Effect of N: Low FER (A: Analytical, S: Simulation).



(a) 3: High FER and short burstiness (b) 4: High FER and long burstiness

Fig. 5. Effect of N: High FER (A: Analytical, S: Simulation).

Therefore, the effectiveness of differentiation (i.e., phase II in TP-LDA) between transmission error failure and collision failure is not significant. Therefore, E-TFRC and TFRC+Phase I exhibit similar throughputs. From both Figures 4 and 5, it can be found that the analytical results are consistent with the simulation results.

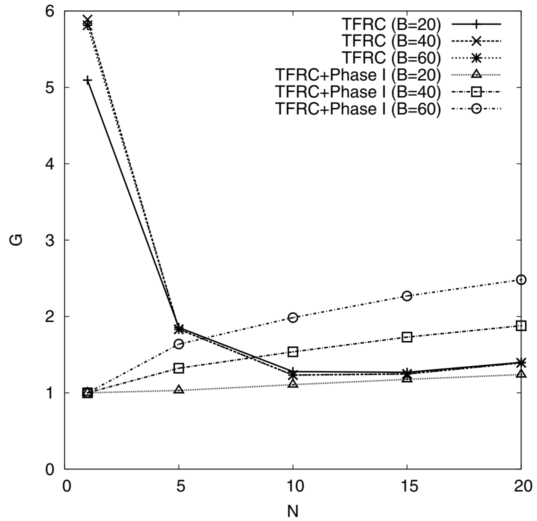
B. Effect of B

To investigate the effect of the AP buffer size B, we define an effective throughput gain G as follows:

$$G = \frac{\text{Effective throughput of E-TFRC}}{\text{Effective throughput of TFRC (or TFRC+Phase I)}}. \quad (24)$$

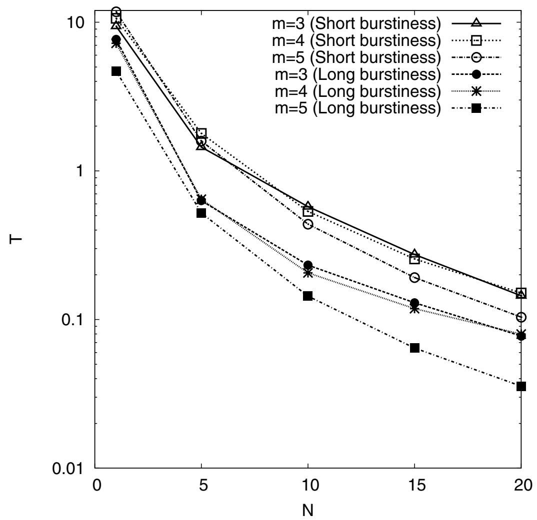
G represents the relative effective throughput of E-TFRC to TFRC or TFRC+Phase I. Therefore, if E-TFRC has a higher effective throughput than TFRC or TFRC+Phase I, G is larger than 1.0.

When the AP buffer size is set to a sufficiently large value, congestion loss is not significant compared to the link loss. In other words, even though B is further increased, the packet loss probability is not drastically affected. As shown in Figure 6, the effect of B is insignificant in TFRC for low FER. On the other hand, in TFRC+Phase I, the link loss is excluded when the packet loss rate is computed, and hence the effective throughput of TFRC+Phase I is directly affected by the congestion loss and the AP buffer size. Specifically, as B increases, congestion losses are drastically reduced and then the link loss becomes a dominant factor to decide the throughput. Consequently, loss differentiation for a link loss (i.e., phase II) is critical for significant improvement in the effective throughput. It can be seen that the throughput gain of E-TFRC over TFRC+Phase I increases with the increase in B. For high FER, we have similar observations and thus the results are omitted due to space limitation.



(a) 1: Low FER and short burstiness (b) 2: Low FER and long burstiness

Fig. 6. Effect of B: Low FER.



(a) Low FER (b) High FER

Fig. 7. Effect of m.

C. Effect of m

In IEEE 802.11 WLANs, a link layer retransmission mechanism is supported to conceal packet losses in the wireless link to the transport layer. Hence, the retransmission limit m affects the packet loss probability and throughput. Figures 7 show the effective throughput of E-TFRC with different retransmission limits. When FER is low, the highest throughput can be achieved with $m = 3$ in most cases (Figure 7(a)). Namely, a small m is better for improving the effective throughput if the channel condition is not poor. For $m = 5$, even though further reduction of the packet loss rate can be achieved, more collisions and additional delay are introduced. On the other hand, if FER is high, more link layer retransmissions are necessary to maximize the effective throughput. Specifically, as shown in Figure 7(b), the effective throughput for channel profile 3 is the highest when $m = 4$. Also, for channel profile 4, $m = 5$ shows a comparable throughput to that for $m = 4$. It can be concluded that there exists an optimal m that maximizes the throughput given N and channel condition, and the throughput can be improved by choosing the optimal m adaptively.

VI. Related Work

To support multimedia streaming applications in the wired Internet, extensive studies have been conducted and several TCP-friendly congestion control schemes have been introduced. However, in the wireless domain, due to severe impairments, the congestion control schemes developed for wired networks meet great challenges. To overcome this problem, several schemes have been proposed such as equation-based schemes [27], [28], additive increase multiplicative decrease (AIMD)-based schemes [29], *etc*. On the other hand, transport protocols for satellite networks and heterogeneous wireless networks have been introduced in [30] and [31], respectively. Shen *et al.* evaluate the TFRC performance over wireless fading channels using a discrete-time Markov model [32].

In [33]- [36], TFRC-based rate control schemes for multimedia traffic in IEEE 802.11 WLANs have been developed.

Borri *et al.* [33] evaluate the performance of TFRC in an experimental IEEE 802.11g WLAN testbed. They observe undesirable throughput degradation and unfairness between TFRC and TCP flows. To remedy this problem, they propose a tuning scheme using *NoFeedback timer* and *Backoff parameter*. They present valuable experimental results, but the effects of the IEEE 802.11 MAC protocol are not fully investigated.

Wang *et al.* [34] introduce an adaptive buffer rate control (ABRC) scheme. In ABRC, an arbitrary bandwidth allocation for TCP and UDP flows is achieved for smooth multimedia transmission. The buffer occupancy is estimated and the sending rate is adjusted. Since ABRC sets the sending rate regardless of packet loss events, ABRC is not sensitive to transmission errors in a wireless link. However, the packet losses due to channel contention in IEEE 802.11 WLANs are not considered.

Li *et al.* [35] evaluate TFRC over wireless ad-hoc networks. They focus on retransmission and backoff behaviors in IEEE 802.11 WLANs. To avoid setting an inaccurate sending rate, the sender determines the sending rate depending on the measurement results and the model value for round trip time in IEEE 802.11 wireless ad-hoc networks. Fu *et al.* [36] describe the design of a transport protocol for multimedia transmission for MANETs. Specifically, they develop a multi-metric joint identification scheme for packet and connection behaviors based on end-to-end measurements. References [35] and [36] mainly focus on mobility and transmission errors in a wireless link, which are important in MANETs. However, the distinction between transmission failures due to transmission errors and collisions is not addressed. Also, since wireless ad-hoc networks are assumed, the effect of the AP bottleneck is not explored.

Performance studies of TCP and TFRC over IEEE 802.11 WLANs are reported in [37], [38], based on simulation results. Nahm *et al.* [37] examine the steady-state TCP behavior and utility of TFRC in IEEE 802.11 multi-hop networks. Their results indicate that the original TCP window mechanism is inefficient and hence they propose a new window scheme called fractional window scheme which resembles the stop-and-go protocol. In addition, they show that TFRC based on the steady state throughput of TCP in wired networks is not TCP-friendly in IEEE 802.11 multi-hop networks. Similarly, Fu *et al.* [38] investigate the impact of IEEE 802.11 WLANs on the TCP performance. Specifically, they demonstrate that there exists an optimal window size achieving the highest TCP throughput.

In summary, to the best of our knowledge, little work on rate control for multimedia streaming, focusing on the effects of channel collision and the AP bottleneck in infrastructure-based WLANs, has been reported. In addition, no analytical models for studying the TFRC performance in IEEE 802.11 WLANs are available. Therefore, the analytical model for TFRC in IEEE 802.11 WLANs and investigation on the effects of channel collision and the AP bottleneck in different environments presented in this paper should provide useful guidelines and insights.

VII. CONCLUSION

In this paper, we have proposed an enhanced TFRC (E-TFRC), which improves the throughput of TFRC by incorporating a novel loss differentiation algorithm called TP-LDA. In E-TFRC, the receiver differentiates congestion losses from link losses. The ratio of collision failures for a link loss is estimated by a cross layering approach and the ratio is used to update the packet loss event rate. Analytical and simulation results demonstrate that, by accurately estimating packet loss event rate due to congestion, E-TFRC can improve the effective flow throughput, especially when a large number of nodes compete for channel access. In addition, the effects of AP buffer size and retransmission limit are investigated, and the results can be utilized for the performance optimization of TFRC in IEEE 802.11 WLANs. TP-LDA can also be applied to other transport layer protocols, and the investigation of the performance enhancements in TCP and wireless application protocol (WAP) via TP-LDA is under way.

APPENDIX I
CALCULATION OF $E[slot]$

The average slot length $E[slot]$ is given by [18],

$$E[slot] = (1 - P_{tr})\sigma + P_{tr} P_S T_S + P_{tr}(1 - P_S) T_C,$$

where σ is the time duration of an idle slot. $P_{tr} = 1-(1-\tau)^n$ is the probability that there is at least one transmission in a given virtual time slot. P_S is the probability that the transmission is successful and is calculated as $P_S = \frac{n\tau(1-\tau)^{n-1}}{1-(1-\tau)^n} \cdot \pi_g^{WLAN}$, where π_g^{WLAN} is the steady state probability of a good WLAN link condition. T_S and T_C are time durations that the WLAN link is sensed busy during a successful packet transmission and during a collided frame transmission, respectively, and they are given by $T_S = DIFS+H+P+\delta+SIFS+ACK+\delta$ and $T_C = DIFS+H+P+ACK$, where δ is the propagation delay. $DIFS$ and $SIFS$ represent DCF inter-frame space and small inter-frame space, respectively. H, P, ACK are respectively the transmission times for the header, payload, and ACK frame.

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